IN THE UNITED STATES DISTRICT COURT
FOR THE EASTERN DISTRICT OF TEXAS
MARSHALL DIVISION

SAINT LAWRENCE COMMUNICATIONS LLC,	§	
	§	
Plaintiff,	§	**Case No. 2:16-cv-00082-JRG**
	§	
v.	§	**Jury Trial Demanded**
	§	
APPLE INC., AT&T MOBILITY LLC, and CELLCO PARTNERSHIP D/B/A VERIZON WIRELESS,	§	
	§	
Defendants.	§	

DECLARATION OF TOKUNBO OGUNFUNMI, Ph.D.

1. My name is Tokunbo Ogunfunmi. I am offering this declaration in the matter listed above on behalf of Saint Lawrence Communications LLC ("SLC") and at the behest of their attorneys Ahmad, Zavitsanos, Anaipakos, Alavi & Mensing P.C. I am being compensated at my usual rate and my compensation is not dependent on any opinions that I may take in this matter, any testimony, or any intermediate or final resolution in the matter.

2. I have been asked to review the patents in suit in this matter, i.e. U.S. Patents Nos. 6,795,805 (`805 Patent); 6,807,524 (`524 Patent); 7,151,802 (`802 Patent); 7,260,521 (`521 Patent); and 7,191,123 (`123 Patent) (collectively, "the Asserted Patents"). My review of these patents followed my ordinary practice, i.e., I began by reading the patents themselves and then reviewed their respective prosecution histories.

3. My curriculum vitae and testimony list are included in Appendix A to this report. To summarize my qualifications, I hold three academic degrees in the field of Electrical Engineering: A Bachelor of Science and Engineering, a Masters of Science and Engineering, and a Doctorate of Philosophy degree. I received my Bachelor degree (First Class Honors) from University of Ife in Nigeria and my Masters and Doctorate degrees from Stanford University. I have expertise in digital and adaptive signal processing and communications applications of signal processing as well as speech coding. A more comprehensive description of my expertise is reflected in my CV which is attached as Appendix A to this report.

Legal Understanding

4. I am not an attorney. Therefore, I have relied upon certain legal principles that have been explained to me.

5. I have been informed that when interpreting claim terms one must endeavor to employ the perspective of a person of ordinary skill in the relevant art at the time of invention. When interpreting the claims, I understand that the ordinary meaning of the language within the claims should be utilized unless the specification or prosecution history clearly provides reason for applying a different interpretation. In other words if a claim term has a well-known and understood meaning to a person of ordinary skill in the art that meaning should be applied to the claims unless the patentee states a clear and unambiguous alternative meaning that they intend to be in force.

6. I have approached the meaning of the claim terms as one of ordinary skill in the art at the time of the effective filing date of the patents in suit or the patents to which any of them claim priority. This approach puts the priority date for the`123 Patent at November 1999 and the priority date for the remaining Asserted Patents at October 1998. I am currently unaware of any arguments that any of the patents should be allotted an earlier priority date. Therefore, I have no opinion at this moment on any such possible argument.

7. I understand that a number of factors should be considered in determining the level of ordinary skill in the art, including: (1) the educational background of those actively working in the field at the time the invention was made (and particularly of any person(s) who may have independently made the invention(s) at about the same time as the inventor(s)); (2) the type of problems encountered in the art; (3) the various ways that others sought to solve the existing problems; (4) the rapidity with which innovations were being made in the art at the time the invention was made; (5) the level of technological sophistication at the time the invention was made; (6) the educational level of the inventor; and (7) teachings and disclosures of any references that, while not prior art to the invention(s), nonetheless contain teachings or disclosures of what the level of ordinary skill in the field may have been at the time of the invention(s).

8. In my opinion, a person of ordinary skill in the art in October 1998 or November 1999 (“relevant time frame”) would have had a Bachelor’s degree in electrical engineering or computer science and at least 2 years of experience in digital signal processing or speech processing, or would possess equivalent education or experience. In some cases a greater degree of education could compensate for a lesser degree of work experience. Similarly, in some cases a greater degree of work experience could compensate for a lesser degree of education. When I use the term “person of ordinary skill in the art,” I am referring to such a person.

9. I understand that a patent includes a specification that describes the patented invention. The specification includes a written description of the preferred embodiment(s) of the invention, drawings, and claims that define the scope of the patent.

10. I also understand that patents may include dependent and independent claims. Dependent claims include each and every element of the claim or claims from which they depend.

11. I have further been informed that a patent is invalid for indefiniteness if its claims, read in light of the specification delineating the patent, and the prosecution history, fail to inform, with reasonable certainty, those skilled in the art about the scope of the invention.

12. I have been informed that a claim element may be expressed as a means or step for performing a specified function without reciting the specific structure, material or acts in support thereof. I have also been informed that in such instances, the claim element is construed to cover the corresponding structure, material or acts described in the specification and equivalents thereof. I have further been informed that with respect to computer-implemented inventions, the structure disclosed in the specification needs to be more than simply a general purpose computer or microprocessor. Specifically, I have been informed that with respect to such computer-implemented inventions, the specification must disclose an algorithm for performing the claimed function. Further, I have been informed that the patentee may express that procedural algorithm in any understandable terms including as a mathematical formula, in prose, or as a flow chart, or in any other manner that provides sufficient structure.

13. I will attempt to apply these legal principles, as they have been explained to me, in my analysis of the Asserted Patents.

Technology Background

14. The Asserted Patents are directed to improvements to both the decoder and the encoder side of a speech codec which facilitate transmission of wideband speech signals. Specifically, the Asserted Patents are directed to an audio compression model that is particularly effective at modeling human speech. This model (referred to hereinafter as the "Human Speech Model") uses three major components: a fixed codebook, an adaptive codebook, and an LP synthesis filter to model the three main anatomical components of the human speech system: the lungs (1), the larynx (2) and the vocal tract (3) as shown in Figure 1, below.

15. In human speech, the lungs are used to push air up through the trachea. The fixed codebook component essentially models the flow of air coming through the lungs. This air then passes through the larynx which contains the vocal chords. If the sound being made is a voiced sound, such as a vowel—"A" or "E" or "I"—the vocal chords vibrate. If the sound being made is an unvoiced sound, such as the fricative sound formed by the letters "s" and "h"—"SHHHHH"—the vocal chords do not vibrate. The adaptive codebook component essentially models the periodic vibration of the vocal chords. The flow of air output by the larynx then goes through the vocal tract, where it is amplified and given a specific sound characteristic by the shape of the vocal tract components, such as the position of the tongue and the opening of the lips. The LP synthesis filter component essentially models the shape of the vocal tract. Together, the fixed codebook, the adaptive codebook and the LP synthesis filter form the basis for the audio compression model described in the Asserted Patents.

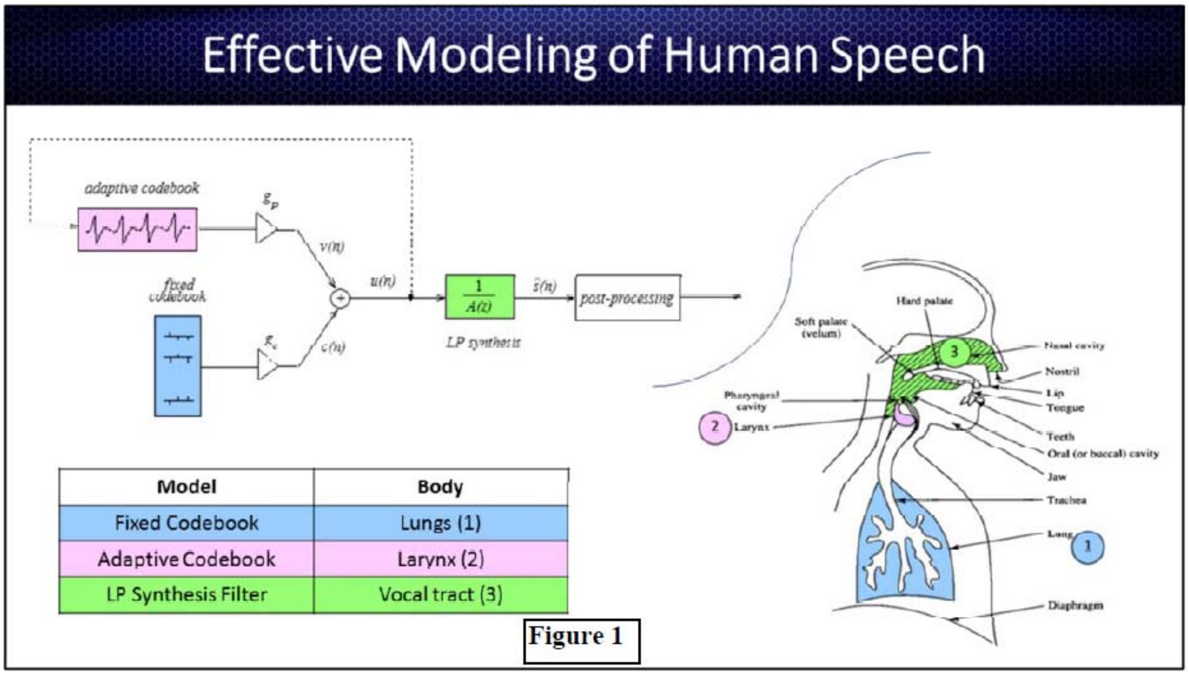

Figure 1

16. Although this model allows an encoder to efficiently model human speech, speech codecs implementing this model face certain problems, particularly in applications involving wideband speech. The Asserted Patents seek to solve these problems.

Meaning of "Wideband [Speech] Signal "

17. The term "wideband" was a term of art in the relevant time frame which was intended to provide a contrast to the term "narrowband" which was used previously to refer to traditional telephone applications that filtered a speech signal in the range of 200-3400 Hz.

18. Due to practical considerations associated with processing speech signals, it was well known to those of ordinary skill in the art that there is not one specific frequency at which a signal goes from being a narrowband speech signal to a wideband speech signal. Instead, there was a general understanding that a wideband speech signal has a bandwidth that is approximately twice as wide as that of a narrowband speech signal.

19. The Asserted Patents acknowledge the demand for "efficient digital wideband speech/audio encoding techniques." [`805 Pat., 1:12-17]. Further, the term "wideband speech" signal was well-known to those of ordinary skill in the art in the relevant time frame and was commonly used as shown, for example, in:

a. P. Mermelstein, "G.722, A new CCITT Coding Standard for Digital Transmission of Wideband Audio Signals," *IEEE* Comm. Mag., Vol. 26, No. 1, pp. 8-15, Jan. 1988 (describing a standard applicable to wideband signals and discussing the frequency range of wideband audio signals compared to narrowband audio signals) (attached as Ex. A);
b. Fuemmeler et. al, "Techniques for the Regeneration of Wideband Speech from Narrowband Speech," EURASIP Journal on Applied Signal Processing 2001:0, 1-9 (Sep.

2001) (noting that some work has already been done in the area of wideband speech regeneration) (attached as Ex. B);

c. C.H. Ritz et. al., "Lossless Wideband Speech Coding," 10th Australian Int'l. Conference on Speech Science & Technology, p. 249 (Dec. 2004) (noting that wideband speech refers to speech sampled at 16 kHz and acknowledging existing research into wideband speech coding) (attached as Ex. C).
d. U.S. 5,581,652, filed Sep. 29, 1993 (titled: "Reconstruction of wideband speech from narrow band speech using codebooks) (Ex. D);
e. U.S. 6,615169, filed Oct. 18, 2000 (titled: "High frequency enhancement layer coding in wideband speech codec") (Ex. E).

20. The AMR-WB Standard implemented by the 3GPP used the term "wideband" in its title further evidencing the fact that the meaning of that term was well-known to those of ordinary skill in the art.

21. The Asserted Patents further make it clear that the wideband [speech] signal discussed therein is not limited to a signal having a bandwidth with a strict cut off at 50-7000 Hz and can include frequency components above 7000 Hz or, alternatively, may not have components all the way up to 7000 Hz:

a. The `805 Patent discloses generating a noise signal in the frequency range of 5.6-7.2 kHz that is then added to the synthesized speech signal to form the wideband speech signal at the output which would include frequencies in the range of 7000 to 7200 Hz. [`805 Pat., 17:64-18:4].
b. The `802 Patent claims recite a decoder for producing a synthesized wideband signal where the bandpass filter has a frequency bandwidth located between 5.6 kHz and 7.2 kHz. [`802 Pat., 21:10-13].
c. The 802 Patent describes an embodiment where the input wideband signal is down-sampled from 16 kHz to 12.8 kHz, reducing "the number of samples in a frame, the processing time and the signal bandwidth below 7000 Hz." [`802 Pat., 2:48-51; claim 1 ("a wideband signal previously down-sampled during encoding")]. This is due to the sampling theorem according to which the highest frequency component in a signal is equal to half of the sampling rate. Accordingly, if a signal is sampled at 8000 samples per second, the maximum frequency component of the resulting signal will be at 4 kHz. In contrast, when a signal is sampled at 16,000 samples per second, the highest frequency component of the resulting signal will be at 8 kHz. Therefore, for the down-sampled wideband signal discussed in the `802 Patent which is down-sampled from 16 kHz to 12.8 kHz, the highest frequency component would be at 6400 Hz, i.e., below 7000 Hz.
d. The `524 Patent defines the output of the preemphasis filter (103) (i.e., signal "S"), as "the wideband signal input speech vector (after down-sampling, pre-processing, and preemphasis)." [`524 Pat., 7:2-3].The `524 Patent further discloses that in the downsampling module (101) the signal is down-sampled "from 16 kHz down to 12.8 kHz," corresponding to a maximum frequency of 6400 Hz. [`524 Pat., 7:45-48]. Accordingly, the `524 Patent discloses a wideband signal with a frequency range less than 7000 Hz.

22. For the reasons discussed above, a person of ordinary skill in the art in the relevant time frame would understand the meaning of the term wideband [speech] signal. Moreover, a person of ordinary skill in the art in the relevant time frame would understand that a wideband speech signal does not necessarily have to include components of speech in the whole frequency range of 50-7000 Hz. Accordingly, from the perspective of a person of ordinary skill in the art in the relevant time frame, a wideband speech signal may have components that go beyond 7000 Hz or, alternatively, may not have components all the way up to 7000 Hz.

Meaning of "Low Frequency Portion"

23. The term "low frequency portion" as used in the claims of the `805 Patent informs a person of skill in the art of the scope of the claims with reasonable certainty.

24. Independent claim 1[1] of the `805 Patent recites "an innovation filter for filtering the innovative codevector in relation to said periodicity factor to thereby reduce energy of a low frequency portion of the innovative codevector and enhance periodicity of a low frequency portion of the excitation signal."

25. It is common for those of ordinary skill in the art to refer generally to "low frequency portion" and/or "high frequency portion" of a signal without specifying explicit cut off frequencies to delineate where such portions begin or end. Specifically, a person of ordinary skill in the art would understand with reasonable certainty what a "low frequency portion" of a signal is based on the context in which that term is used. This is evident, for example, from:

a. U.S. Pat. No. 8,494,667 (Title: "Apparatus for encoding and decoding audio signal and method thereof") ("In accordance with the present invention, the generation of the arbitrary downmix gain (ADG) is carried out in such a manner that the low frequency portion of the ADG is not generated as a gain, but generated by executing residual coding for the low frequency component of the first downmix signal, and the high frequency portion of the ADG is generated as a gain, as in a conventional method, in order to enable the generated ADG to exhibit an improved performance." at 15:10-17) (attached as Ex. F);
b. U.S. Pat. No. 7,991,495 (Title: "Method and apparatus for processing an audio signal") ("Yet, although the decoding of the extension signal is executed, the decoding can be performed on a predetermined low frequency portion of the extension signal only (1450). For instance, there is a case that since the decoding apparatus is a low power decoder, if the extension signal is entirely decoded, efficiency is degraded, or since the decoding apparatus is unable to decode the entire extension signal a predetermined low frequency portion of the extension signal is usable." 12:61-13:2) (attached as Ex. G);

26. In the context of the `805 Patent, the term "low frequency portion" is used in relation to the output of the innovation filter and would have been understood by a person of ordinary skill in the art.

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[1] Although my analysis focuses on independent claim 1, the same reasoning is applicable to the remaining claims in which this term is in dispute.

27. One limitation of the Human Speech Model results from a combination of two factors: (1) audible frequencies in wideband speech extend to very low values (e.g., down to 50 Hz) where the speech signal has high spectral dynamics and the hearing system is highly sensitive to noise between the harmonics of speech (i.e., the peaks in the spectrum of voice speech); and (2) speech compression using the Human Speech Model is lossy (i.e., noise is introduced in the decoded speech) and although this noise may be dynamically shaped at the encoder, it still extends over the whole audible frequency band of the wideband speech signal (i.e., about 50-7000 Hz).

28. The figure below is an illustrative example of a typical wideband speech signal in a voiced segment in the frequency domain, along with an illustration of the noise introduced by the lossy encoder/decoder pair.

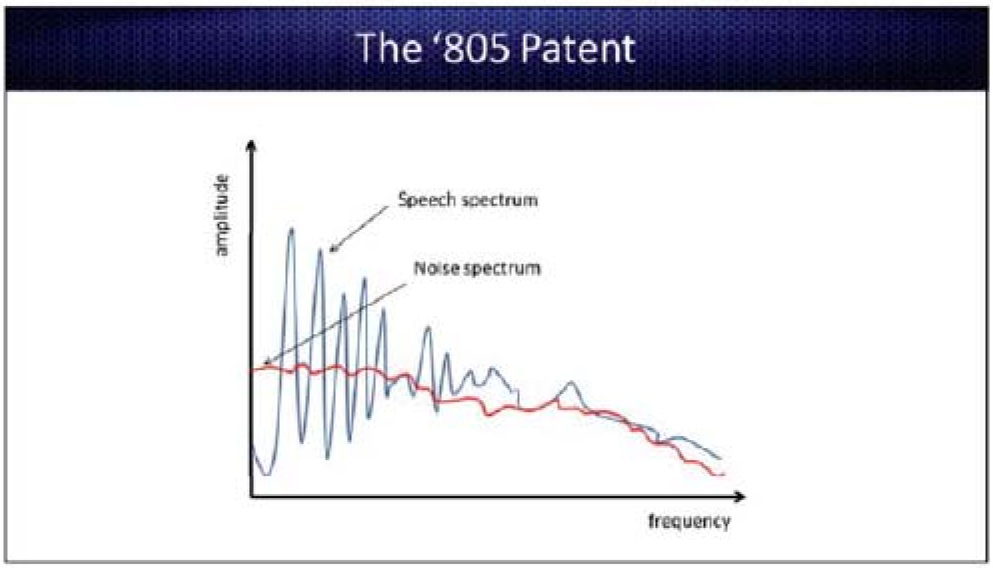

Figure 2

29. As shown in the illustrative figure (Figure 2) above, at the lower frequencies, the relation between the peak of a harmonic and the "valley" between two harmonics is larger compared to the higher frequencies. Accordingly, any noise can have a highly adverse impact in the lower frequency part of the speech spectrum. Moreover, since typically noise levels increase with a decrease in the bit rate, this problem becomes increasingly apparent at lower bit rates. The `805 Patent proposes a solution to this problem.

30. The `805 Patent discloses a solution whereby a decoder at the receiver modifies the content of the fixed codebook as a post processing step. The encoder is not affected by this post processing step which is only applied at the decoder. Since an encoder/decoder pair using the Human Speech Model introduces noise in the signal, this noise is interpreted as originating from the fixed codebook. The noise is due to the analysis-by-synthesis procedure where the decoder is embedded in the encoder. The approximation error resulting from the difference between the "best entry" from the fixed codebook and the actual portion(s) of the speech signal not modeled by the LP synthesis filter and the adaptive codebook is the coding noise.

31. Specifically, at the encoder of the transmitter, the first step is to calculate the parameters of the LP synthesis filter. The second step is to calculate the adaptive codebook (also referred to as "pitch codebook") parameters and the third step is to identify the best entry from the fixed codebook (also referred to as "innovative codebook"). The identification of the best entry from the fixed codebook is an approximation in the sense that it aims to identify the entry which best models, without being equal to, the portion(s) of the speech signal that the LP synthesis filter and the adaptive codebook were not able to model. The approximation error resulting from the difference between the "best entry" from the fixed codebook and the actual portion(s) of the speech signal not modeled by the LP synthesis filter and the adaptive codebook is the coding noise.

32. One way to reduce the coding noise is to have an encoder that makes a better approximation when identifying the "best entry" from the fixed codebook. However, the availability of a limited number of bits (depending on the bit rate) limits the encoder's ability to improve this approximation. Another solution is to increase the bit rate, which effectively, increases the number of available possibilities for the "best entry" in the fixed codebook. However, the bit rate available to the encoder is constrained by the transmission network.

33. The `805 Patent proposes a third solution which decreases the perception of the coding noise in a post-processing step at the decoder. To that end, the low frequency content of the fixed codebook is reduced in proportion to the periodicity of the decoded signal, i.e., the higher the periodicity of the decoded signal, the more attenuation is applied to the low frequencies of the fixed codebook. Although the proposed solution reduces the optimality of the fixed codebook entry selected by the encoder, this reduction in optimality only occurs in the low frequencies and only when the decoded signal is mostly periodic. Therefore, the solution proposed by the `805 Patent significantly reduces the possibility of perceiving unwanted coding noise in the low frequencies without a significant impact on the decoded signal in any other frequency band.

34. Accordingly, the `805 Patent discloses calculating a periodicity factor for the wideband speech signal at the decoder (i.e., the excitation signal "u" in the `805 Patent) and filtering the selected entry (innovative codevector) from the fixed codebook (i.e., innovative codebook) through an innovation filter (205 -- F(z)) whose coefficients are calculated according to the periodicity factor of the wideband speech signal at the decoder.

35. First, a voicing factor rv is computed in the voicing factor generator (204) as follows:

$$\boldsymbol{rv} = \frac{\boldsymbol{Ev} - \boldsymbol{Ec}}{\boldsymbol{Ev} + \boldsymbol{Ec}}$$

where Ev is the energy of the scaled pitch codevector bvT (i.e., the decoded, scaled codevector v(n) from the adaptive codebook of the Human Speech Model); and Ec is the energy of scaled innovative codevector gck (i.e., the decoded, scaled codevector c(n) from the fixed codebook of the Human Speech Model). Accordingly, if the decoded signal exhibits a speech like periodic nature (i.e., if Ev is much larger than Ec) the ratio rv will be close to 1 and if the decoded signal is noise like (i.e., if Ec is much larger than Ev) the ratio rv will be close to -1. The ratio rv is between -1 and +1.

36. A periodicity factor (α) is then calculated as:

$$\alpha = 0.125(1 + rv)$$

37. Accordingly, α is close to 0.25 if the decoded speech signal exhibits a speech like periodic nature and α is close to 0 if the decoded signal is noise-like and not periodic.

38. An innovation filter is then constructed as a linear, 3-tap, symmetrical finite impulse response (FIR) filter with the transfer function:

$$F(z) = -\alpha z + 1 - \alpha z^{-1}$$

39. The frequency response of the innovation filter (205 – F(z)) is therefore shaped based on the periodicity factor and emphasizes the higher frequencies more than the lower frequencies. Since α is positive or equal to 0, the innovation filter F(z) is always a high pass filter (i.e., attenuates low frequency components and lets the higher frequency components pass through), with a stronger attenuation of the low frequencies when α is largest (i.e., when the decoded wideband speech signal is more periodic or voiced).

40. The selected entry (innovative codevector) from the fixed codebook (i.e., innovative codebook) is filtered through F(z). The coefficients of the innovation filter (F(z)) are calculated such that this filter will reduce the energy of the low frequency portion of the innovative codevector (the entry selected from the fixed codebook) and thereby enhance the periodicity of the total excitation signal (i.e., sum of the contributions from the fixed codebook and the adaptive codebook). This reduction in the intensity of the coding noise in the low frequencies is shown in the illustrative figure below (Figure 3):

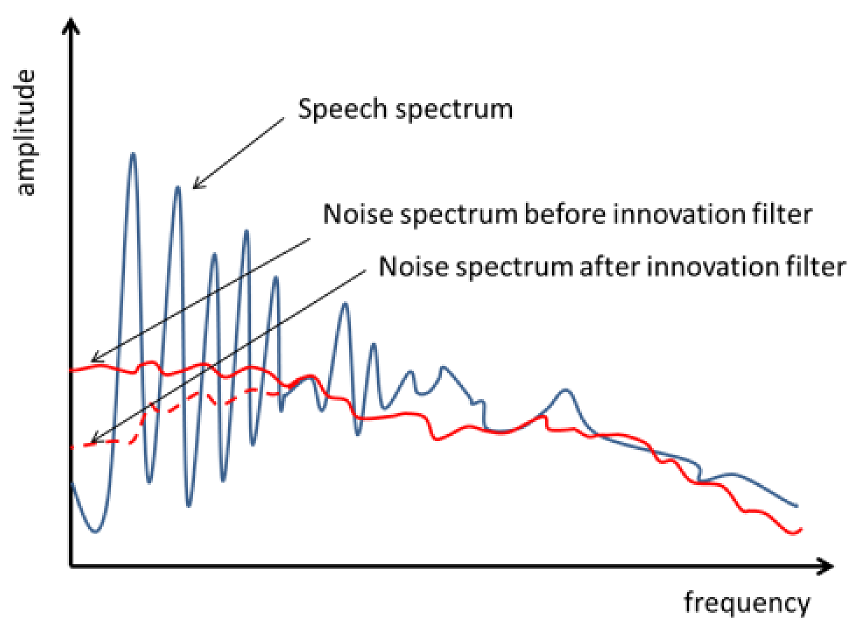

Figure 3

41. Accordingly, the proposed solution of the `805 Patent reduces the perceived distortion at the receiver and improves the quality for the listener. Moreover, most of the energy of the harmonics (peaks in the speech spectrum) in voiced speech comes from the adaptive codebook. Therefore, the innovation filter (F(z)) will not have much impact on the harmonic peaks of the speech signal. Instead, as shown in the figure above, the innovation filter (F(z)) mostly affects (i.e., reduces) the noise level at low frequencies.

42. The term "low frequency portion" is used to describe the result of applying the innovation filter. Further, in the context of the invention disclosed in the `805 Patent (as described above), from the perspective of a person of ordinary skill in the art in the relevant time frame the meaning of the term "low frequency portion" and the claim scope would have been understood with reasonable certainty.

Meaning of "High Frequency Content"

43. The term "high frequency content" as used in the claims of the `524 Patent informs a person of skill in the art of the scope of the claims with reasonable certainty.

44. Independent claim 1[2] of the `524 Patent recites "a signal pre-emphasis filter responsive to the wideband speech signal for enhancing a high frequency content of the wideband speech signal to thereby produce a pre-emphasized signal."

45. It is common for those of ordinary skill in the art to refer generally to "low frequency portion" and/or "high frequency portion" of a signal without specifying explicit cut off frequencies to delineate where such portions begin or end. Specifically, a person of ordinary skill in the art would understand with reasonable certainty what a "high frequency content" of a signal is based on the context in which that term is used. This is evident, for example, from:

a. U.S. Pat. No. 8,494,667 (Title: "Apparatus for encoding and decoding audio signal and method thereof") ("In accordance with the present invention, the generation of the ADG is carried out in such a manner that the ***low frequency portion*** of the ADG is not generated as a gain, but generated by executing residual coding for the low frequency component of the first downmix signal, and the ***high frequency portion*** of the ADG is generated as a gain, as in a conventional method, in order to enable the generated ADG to exhibit an improved performance." at 15:10-17) (attached as Ex. F);
b. U.S. Pat. No. 7,991,495 (Title: "Method and apparatus for processing an audio signal") ("Yet, although the decoding of the extension signal is executed, the decoding can be performed on a predetermined ***low frequency portion*** of the extension signal only (1450). For instance, there is a case that since the decoding apparatus is a low power decoder, if the extension signal is entirely decoded, efficiency is degraded, or since the decoding apparatus is unable to decode the entire extension signal a predetermined ***low frequency portion*** of the extension signal is usable." 12:61-13:2) (attached as Ex.G);

[2] Although my analysis focuses on independent claim 1, the same reasoning is applicable to the remaining claims in which this term is in dispute.

46. In the context of the `524 Patent, the term "high frequency content" is used in relation to the output of the innovation filter and would have been understood by a person of ordinary skill in the art.

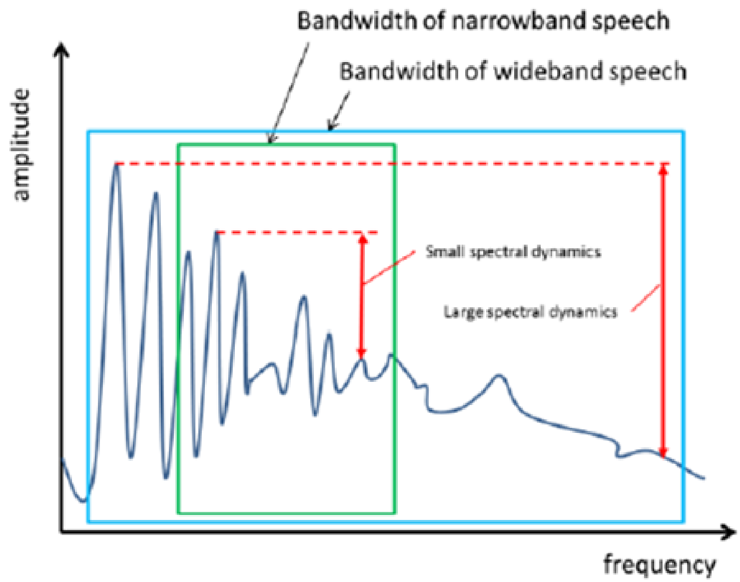

Figure 4

47. The function of the pre-emphasis filter (103) is to enhance the high frequency contents of the input speech signal. [`524 Pat., 8:9-10]. Specifically, as discussed above, the spectrum of a wideband speech signal typically exhibits much larger spectral dynamics (i.e., ratio of maximum value to minimum value in the spectrum) compared to a narrowband speech signal. Figure 4 (above) is an example to illustrate the difference in spectral dynamics of a wideband speech signal and a narrowband speech signal.

48. Accordingly, when preprocessing a wideband speech signal to be encoded, it is desirable to reduce the spectral dynamics of the signal, in particular for fixed point implementations (e.g., in cellular phones) where a limited resolution (i.e., number of bits) is permitted for representing the data. In signals with large spectral dynamics, implementation of arithmetic with limited resolution can result in inefficient calculations, especially for the LP synthesis filter coefficients. The encoder thus operates in a pre-emphasis domain by enhancing the high frequency content of the input speech signal which results in a reduction in the dynamic range of the input speech signal. [`524 Pat., 8:9-14].

49. One of ordinary skill in the art would understand that the specific frequency range for what constitutes this "high frequency content" is not necessary to understand the claim scope with reasonable certainty. Instead, the term "high frequency content" is intended as a term to describe this content in contrast to the low frequency content of the speech signal which typically has a higher amplitude. In fact, the specific cut off frequency for the "high frequency content" relative to a low frequency content is in part, a function of the particular speech sound being transmitted. Therefore, one of ordinary skill in the art would understand that it would not be reasonable to define the term "high frequency content" in terms of specific frequency cut offs.

Instead, one of ordinary skill in the art would understand that the use of a relative term, i.e., "high frequency content" is appropriate to describe the claimed invention of the `524 Patent.

50. It is my understanding that defendants have not yet submitted an expert declaration regarding the issues discussed in this declaration. Further, it is my understanding that the defendants have not yet submitted a brief to the court explaining the basis for their positions with respect to the various claim terms. Accordingly, I reserve the right to supplement this declaration or otherwise address any arguments that may be raised by any expert retained by the defendants or by the defendants in their briefs.

I declare under penalty of perjury that the foregoing is true and correct.

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TOKUNBO OGUNFUNMI

Executed on Dec. 2, 2016